**Claims**

Sub A1

1. Method for the compression of an electric audio signal which is produced in the process of recording the ambient noise by means of an electroacoustic transducer, more particularly a microphone,
wherein
- the amplitude of said audio signal or of a derived digital or analog signal is normalized to a first predetermined range D;
- said audio signal is mapped using a nonlinear function onto a second predetermined range of values W in order to obtain an emphasis of sensitive value ranges; and
- the result is stored in an electronic memory in a digital form.

2. The method of claim 1, wherein a nonlinear function is used whose slope dW/dD decreases with increasing values in order to obtain an emphasis of the small values of said first range of values.

3. The method of claim 1, wherein said result is represented by binary numbers having a fixed number of binary digits from 3 to 16 bits, preferably from 4 to 8 bits, and more preferably of 4 bits.

4. The method of claim 1, wherein said audio signal is divided into at least two band signals by filtering, each one of the band signals containing a frequency range of the audio signal, and each band signal only containing the content of the other band signals in a clearly attenuated form, more particularly attenuated to the half, or not at all.

Sub A1

5. The method of claim 4, wherein 3 to 15, preferably 4 to 10, more preferably 5 to 8, and particularly preferably 6 band signals are produced.

Sub B2

6. The method of claim 4, wherein said band signals essentially contain frequency ranges of the same width each, and all frequency ranges are comprised in the range of 500 Hz to 10,000 Hz.

Sub A2

7. The method of claim 4, wherein the band signals are generated by a single or a cascaded multiple splitting of an input signal which is the audio signal or one of the output signals in applying the following steps:
- first low pass filtering generating a first output band signal,
- subtraction of the first output band signal from the input signal for the generation of a second output band signal;
all first low pass filterings preferably having the same Q-factor.



8. The method of claim 7, wherein said low pass filtering is realized by means of a digital convolution over 10 - 30 values, preferably 15 - 25 values, and more preferably 19 values.

9. The method of claim 8, wherein for the purpose of the low pass filtering, the convolution is performed with the terms $a_i * x_{t-i}$, the coefficients $a_i$, $0 \leq i \leq 18$, being approximately equal to {0.03, 0.0, -0.05, 0.0, 0.06, 0.0, -0.11, 0.0, 0.32, 0.50, 0.32, 0.0, -0.11, 0.0, 0.06, 0.0, -0.05, 0.0, 0.03}.

10. The method of claim 7, wherein the input signal is digitized and only every nth value of each division stage is added to the band signal, n being at least 2 and preferably

~~n = 2, in order to compensate for the increased data volume resulting from the splitting into band signals.~~

~~11. The method of claim 1, wherein an energy signal which is proportional to the energy content is generated from said audio signal or from a signal derived therefrom, said energy signal preferably being generated by squaring.~~

SUB C5

12. The method of claim 11, wherein said energy signal is subjected to ~~a~~ second low pass filtering.

Sub A3

~~13. The method of claim 12, wherein said second low pass filtering is effected digitally in the form of a convolution over 20 to 70 values, preferably 40 to 55 values, and more preferably 48 values approximately, the coefficients of the convolution preferably being essentially equal to each other and more preferably equal to 1.0.~~

~~14. The method of claim 13, wherein said second low pass filtering is followed by a second data reduction where one energy value among n filtered values is selected, n being at least equal to 2 and preferably equal to the number of values of the convolution of the second low pass filtering.~~

~~15. The method of claim 11, wherein a subsequent differentiation of the energy signal with respect to the time is effected in order to obtain an energy difference signal, said differentiation preferably being effected by computing the difference between each two respective values of the signal.~~

~~16. The method of claim 1, wherein the normalization to a range of values W, which is defined by a lower limit $W_u$, preferably 0, and an upper limit $W_o$, where $W_o - W_u$ is preferably equal to $2^n-1$, n being a whole number greater~~

than 4 and preferably equal to 7, is effected by:
- obtaining the maximum of the absolute value of the input signal within the normalizing duration of the signal, which is shorter or preferably equal to the duration of a hearing sample,
- by multiplying the reciprocal value of said maximum by ($W_0$ - $W_u$ + 1), and
- by multiplying this product by each value of the input signal within the duration of the normalized signal.

17. The method of claim 1, wherein essentially all steps of the method are performed by integer or fixed point arithmetic, preferably by binary arithmetic with a number of digits as provided by the employed computing unit.

18. Device for carrying out the method of claim 1, wherein the device includes a hearing sample unit comprising at least one signal processor which memory is destined to perform at least one processing step of the method.

19. The device of claim 18, wherein a non-volatile semiconductor memory is connected to said processor which allows to store the results of the method.

20. The device of claim 18, wherein a timer is connected to the power supply of said hearing sample unit which allows to switch off the hearing sample unit when no processing activity is required, more particularly in the periods between the processing of two hearing samples, in order to reduce the energy consumption.

21. The device of claim 20, wherein the power supply of said non-volatile memory and/or said memory itself is connected to a timer in such a manner that the memory is essentially capable of being operated only during the

Sub A3

storage of the results in order to reduce the energy consumption by the memory.

22. The device of claim 18, wherein it is in the form of an object which is usually carried by persons, preferably in the form of a wristwatch.

23. Method for the evaluation of the results of the hearing sample processing according to claim 1, wherein program samples of the monitored programs are recorded which have at least the same duration as the hearing samples, the program samples are subjected to the same processing steps as the hearing samples, and a calculation of a first correlation of the hearing samples with the processed program samples is effected in order to find a match.

24. The method of claim 23, wherein the recording of the program samples is started sufficiently before that of the hearing samples and its duration is sufficiently longer than that of the hearing samples to ensure that in the correlation, time shifts between the timer for the hearing samples and the timer for the program samples can be compensated by a displacement in time of the hearing samples with respect to the program samples.

25. The method of claim 23, wherein said first correlation is a standard correlation according to the formula

$$c_t = \frac{\sum_{i=1}^{N}(s_i\, m_{i-t})}{\sqrt{\sum_{i=1}^{N}(s_i)^2}\sqrt{\sum_{i=1}^{N}(m_{i-t})^2}}$$

where

| | | |
|---|---|---|
| N | : | number of values of the hearing sample which are used in the correlation, |
| t | : | time shift |
| $s_i$ | : | hearing sample value at the time i, |
| $m_i$ | : | program sample value at the time i, |
| $c_i$ | : | correlation value for the time shift t: $-1 \leq c_t \leq 1$. |

Sub A4

26. The method of claim 24, wherein the comparison of the hearing samples with the program samples is effected in two passes, a respective hearing sample being compared to all program samples in all ways in the first pass by means of said first correlation whose calculation is simpler due to a coarser graduation of the time shift, while in the case of a time shift whose correlation values $c_t$ are above a predetermined limit, a second, rugged correlation is effected which provides a finer graduation of the time shift and in particular, a time resolution which is at least twice as high as in the first correlation, said second correlation preferably being chosen such that great deviations between the hearing and the program sample have a smaller influence upon the correlation coefficients than in the first correlation, and preferably being effected according to the formula

$$r_t = \frac{\sum_{i=1}^{N} |s_i - a * m_{i-t}}{\sum_{i=1}^{N} |s_i|}$$

where

| | | |
|---|---|---|
| N | : | number of hearing sample values used in the correlation, |
| t | : | time shift between the hearing and the program sample, |

Sub A4

$s_i$ : hearing sample value at the time i,
$m_i$ : program sample value at the time i, and
a : scaling factor which takes account of the damping of the program signal with respect to the hearing sample;
$r_t$ : correlation value for the shift t, 0 (optimal correlation) $\leq r_t \leq 1$ (no correlation),

a being determined in such a manner that $r_t$ assumes a minimal value.

27. Data carrier, more particularly magnetic, optical or magneto-optical data carrier, containing a recorded program upon whose execution the method according to claim 1 is carried out.

28. Data carrier, more particularly magnetic, optical or magneto-optical data carrier, containing a recorded program upon whose execution the method according to claim 23 is carried out.

29. Device comprising at least one program controlled processor unit and a memory for the storage of the program controlling said processor unit, wherein said memory contains a program under whose control at least one and preferably all operations of the method of claim 1 can be performed.

Add B6

- - - - -

ADD C15